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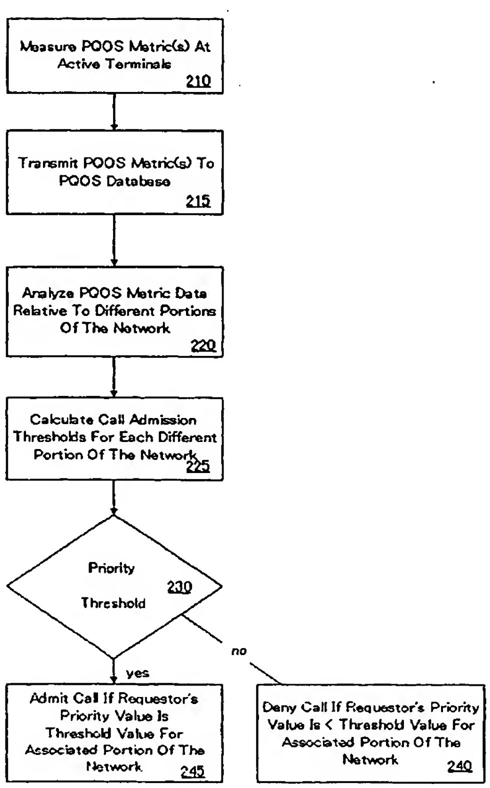
(54) Title: SYSTEM AND METHOD FOR PERCEPTUAL QOS-BASED CALL ADMISSION FOR VOIP, VOIPOW, AND CDMA SYSTEMS



(57) Abstract: ABSTRACT A method and system that utilize perceptual quality of service (QoS) metrics to determine whether to admit new calls onto a VoIP network is described. Perceptual QoS metrics are generated at a communications device, such as a telephone, that represent the measurement of perceptible variations in the quality level of a voice signal, or audio and video signal received at the communications device. A call admission threshold is generated from the perceptual QoS metrics for each node in the network that has communications devices attached thereto. When a request to admit a new call on the network through a node is received, the call admission threshold is compared with a priority value associated with the request to determine whether to admit the new call onto the network.

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DESCRIPTION

SYSTEM AND METHOD FOR PERCEPTUAL QoS-BASED CALL ADMISSION FOR VoIP, VoIPoW, AND CDMA SYSTEMS

(Technical Field)

This invention relates generally to Voice over IP (VoIP) and Mobile communications networks, and more specifically to methods and apparatus for determining whether to admit a new call onto a network based on the perceptual quality of service of other communications active on the network, as well as the required perceptual quality level by the user requesting the new call.

(Background Art)

The use of Internet telephony or voice over IP (VoIP) is rapidly expanding as consumers and businesses alike endeavor to reduce their telecommunications expenses.

In contrast to traditional circuit switched telephone networks wherein each call is given its own individual wire line path between the call participants, VoIP calls are broken down into packets, which are sent with packets from other calls over shared network connections. Based on an internet address contained within each packet, the packet is directed through one or

more routers and switches towards its intended destination. At the intended destination, the packets are reassembled into a continuous voice signal, which is typically played to a recipient through a telephone or another type of voice terminal.

As more and more users utilize the VoIP network simultaneously, various routers and switches on the network may become taxed beyond their capacity causing their packet queues to overflow, and packets to be lost in transport. Furthermore, as a VoIP network becomes taxed near or at capacity the delay in transport of a voice signal increases. The result of both occurrences is a decrease in the perceptual quality of service (QoS) by the users of the network.

In prior art VoIP systems one common method to prevent the loss or unreasonable delay of packets has been to set hard capacity limits at the various network nodes. A network node is typically a switch, router or gateway device that has one or more voice terminals (such as telephones) attached to it. By setting limits at each node, the number of simultaneous callers on the network can be managed to ensure that a minimal number of packets are lost or delayed. Unfortunately, the number of calls accepted at each node is typically not adjustable based on the traffic of the network as a whole, and in certain

circumstances, a node will prevent a new call from being accepted because its hard limit has been reached despite the level of traffic on the VoIP network being sufficiently low to support the additional caller. The end result of utilizing hard limits is an inefficient use of network resources in certain circumstances.

Another rapidly growing area of telephony is the integration of VoIP with wireless communications in VoIP over wireless (VoIPoW) (or mobile communications, such as TDMA or CDMA) networks. The determination whether to admit a call on the wireless network is typically based on the signal to interference ratio (SIR) of the active users of the system. If the SIR is high, a call will be admitted. If the SIR is low, the call will be blocked.

Frame Error Rate (FER) which describes the percentage of data frames that are lost or damaged during transport over the wireless channel is directly related to the SIR. It can be appreciated that if frames are dropped in concentration with each other, significant reductions in a user's perceptual QoS could result. Conversely, if the dropped or damaged frames are distributed throughout the packetized voice signal, they may not significantly impact the resulting perceptual QoS. Accordingly, in certain circumstances, using SIR to make call admittance determinations may result in inefficient

use of network resources; whereas, in other circumstances, calls may be admitted into a network when the perceptual QoS is below acceptable levels.

In VoIPoW networks, the QoS perceived by a user of the network also depends on the manner in which the VoIP packets are compressed to account for the relatively limited bandwidth of the wireless network. By compressing the headers in each VoIPoW packet, the relative percentage of voice data transmitted with each packet is increased. Generally, this is accomplished by transmitting headers that contain contextual information only indicating the changes in the header relative to the previous header. If a header is lost or damaged during wireless transport, the context of subsequent headers are lost as well and the packets are typically discarded until a new uncompressed packet can be sent to re-establish context. Compressed headers allow a greater number of simultaneous calls to be supported; however, poor wireless channels conditions can cause a significant drop in QoS as lost or damaged packets cause a large number of subsequent packages to be discarded because of the loss of context. Uncompressed headers help ensure a high QoS, but because of the high header overhead a much smaller number of calls can be supported over the wireless network at one time.

(Disclosure of Invention)

A method and system for admitting or denying new signals or calls on a network based on the perceptual quality of service realized by active signals or calls is utilized. It should be appreciated that the present invention is applicable to any type of network system in which perceptual QoS is important. Therefore, the following described embodiments are merely illustrative of the advantages of the present invention and are not to be considered an exhaustive list of the invention's applications. For example, the invention is described in great detail with respect to VoIP systems. But the advantages of the invention can intuitively be applied to other types of networks, i.e., data, audio and video, as well.

In a first embodiment, a connection is established between a first receiving device and the network through a node of the network. A packetized voice or audio and video signal is transmitted over the network, and through the node to the receiving device. At the receiving device, the perceptual quality of the received signal is determined in real time by measuring the occurrences of perceptible anomalies within the signal. In variations of the first embodiment, the receiving device may be a

dedicated quality of service test apparatus capable of receiving a data, voice, audio or video signals. The receiving device may also be a communications device such as a telephone or a personal computer. The node may be a router or a gateway terminal. In addition, the node may connect two networks that utilize different transmission technologies. In one variation, an admission threshold value is generated based on the quality level of the signal, and the threshold value is utilized by the node to determine whether to admit a new signal that has been requested onto the network through the node.

In a second embodiment a request to admit a new call is received by a network node. A priority number is associated with the call request. The priority number is compared to an applicable call admission threshold value for the network node through which the new call has been requested, and based on this comparison the new call is admitted onto the network or denied admission. In variations of the second embodiment, the priority number of the call request can be derived from priority numbers assigned to communications devices associated with the call request or the priority number can be assigned to the call request randomly. In a preferred variation the call admission threshold value and the priority number both have numerical values between 0 and 1, and the call

is admitted only if the priority value is greater than the call admission threshold value.

In a third embodiment, a VoIP network comprising a plurality of routers, nodes and communications devices is described. The described components are coupled with each other for the transmission of voice signal packets over the network. At least one communications device is connected to each node, wherein the communications device includes a perceptual QoS agent that measures the occurrence of perceptible variations in a voice stream received at the communications device from the network. The perceptual QoS agent uses the measurements to generate perceptual QoS metrics relative to the node with which it is connected. In variations of the third embodiment, the network may also include a call admissions controller and a perceptual QoS database. The call admissions controller uses the perceptual QoS metrics to generate call admission thresholds for each of the nodes. The perceptual QoS database stores the perceptual QoS metrics information generated by the perceptual QoS agents.

In a fourth embodiment a method for operating a wireless VoIP network is described. First, the presence of perceptible variations in a voice stream received by a wireless device from a base station transceiver are

determined and a perceptual QoS metric is generated based on the variations. Based on the perceptual QoS metric, a call admission threshold is determined for the base station transceiver. In variations of the fourth embodiment, the call admission threshold is utilized to determine whether to admit a new wireless call onto the network through the base station transceiver. In another variation, the perceptual QoS metric is utilized to adjust the network's transmission related resources to maintain an acceptable perceptual QoS level for a wireless device actively participating in a call over the network. In determining whether to admit a new call, the call admission threshold may be compared with a priority value that is assigned to the wireless device involved in the call request.

(Brief Description of Drawings)

The objects, features and advantages of the present invention are readily apparent from the Detailed Description of the Preferred Embodiments set forth below, in conjunction with the accompanying Drawings in which:

Figure 1 is a block diagram illustrating a VoIP network that has been integrated with other telephony networks according to one embodiment of the present invention.

Figure 2 is a flow chart illustrating the operation of the VoIP network in determining whether to admit a new call onto the network according to one embodiment of the present invention.

(Best Mode for Carrying Out the Invention)

The numerous innovative teachings of the present application will be described with particular reference to the presently preferred exemplary embodiments. However, it should be understood that this class of embodiments provides only a few examples of the many advantageous uses of the innovative teachings herein. In general, statements made in the specification of the present application do not necessarily delimit any of the various claimed inventions. Moreover, some statements may apply to some inventive features but not to others.

Further, it should be understood that the present invention is applicable to any type of communication system in which perceptual QoS is important. The following detailed description is merely one embodiment of the invention. Other types of networks the may benefit from the disclosures of the present invention may include, but are not limited to, voice, data, audio and video.

Referring to Figure 1, a voice over internet protocol (VoIP) network 100 is illustrated that can be utilized with embodiments of the present invention. The network 100 typically comprises a plurality of interconnected switches and routers that transfer packets containing voice data to and from various voice terminals according to the IP addresses and information contained within an internet protocol header that is part of each VoIP packet.

Voice terminals, such as Internet ready voice telephones 105 and personal computers 110, may be directly connected to the VoIP network 100 by way of a router 115. Each terminal that is directly connected to the VoIP network has the capability of packetizing voice data for transmission over the network and reassembling packetized voice data received from the network.

Typically, at least one voice terminal of one or more terminals connected to each network node (such as IP router node 115) has a perceptual quality of service agent running in the background to measure the perceptual quality of a voice signal it is receiving and to generate one or more perceptual QoS metrics. In an alternative embodiment, special perceptual measurement terminals 120 that are connected to each node 115 can be utilized to measure quality conditions and generate representative

perceptual QoS metrics for the terminals connected to the node.

Preferably, the perceptual QoS agent produces one or more perceptual QoS service metrics based on variations in the quality of a voice signal that are perceptible when the voice signal is played. Perceptible variations can include but are not limited to voice choppiness, delay variation (jitter), and variations in active speech levels. Software agents for measuring telephone voice quality are commercially available, such as Multi-VQ and Dual-VQ that are sold by Genista Corp. of Japan, the assignee of the present invention. For instance, Dual-VQ is a voice quality tool that is run on a windows platform to analyze voice signals for various numerical or digital characterizations of audio variations that are humanly perceptible. Dual-VQ provides a variety of metrics describing the quality of the analyzed voice signal and has been found to exhibit a high degree of correlation with average voice quality values assigned by people in listening tests. It is within the ordinary level of skill of someone in the software and programming arts to port a version of the voice perceptual QoS tools for operability within voice terminals, especially personal computer based voice terminals 110, for the real time generation of perceptual QoS metrics.

The VoIP network can also be interfaced with other types of networks by way of a gateway (such as a IP-PBX gateway 140). Gateways 140 transform the control and overhead portions of voice signals transmitted from one network into a format that is compatible with another network. For instance, when a telephone call is initiated on a switched line network from a first telephone to a second telephone, a telephone number is used to indicate the line switching necessary to connect the first and second telephones. However, if at least a portion of the call is to be transmitted over a VoIP network 100, a gateway device between the switched network and the VoIP network must translate the phone number (or at least a portion of it) into an IP address, which indicates the VoIP node through which the second telephone is connected to the VoIP network. Furthermore, gateways can transform the voice signal into formats compatible with a particular network. For instance, an analog 64 kbps signal from a switched network can be digitized and compressed by a voice coder/decoder (vocoder) to an 8 kbps digital signal for transmission over a VoIP network.

A portion of a private branch exchange (PBX) network 130 is illustrated in Figure 1. The PBX network 130 comprises one or more telephone voice terminals 135

connected to the gateway 140. The IP-PBX gateway 140 is in turn connected to the VoIP network 100.

The VoIP network of Figure 1 is also connected with a CDMA wireless network 150 by way of an IP-CDMA gateway 155. The IP-CDMA gateway 155 is connected to a mobile switching center (MSC) 160. The MSC 160 controls the flow of voice signals from the gateway 155 and one or more base transceivers 165. Each base transceiver 165 communicates with wireless devices, such as cellular telephones 170 and computers with wireless modems 175 in a geographic area (referred to as a cell) associated with the base transceiver 165 over a defined spread spectrum band of radio frequencies.

Many CDMA devices such as internet ready mobile telephones 170 are capable of decoding and reassembling IP packets. Accordingly, the IP-CDMA gateway 155 is not required to reformat the packetized data into a format that is recognizable by devices connected to the CDMA network. This is in contrast to voice data being sent from a VoIP network gateway to a traditional telephone on a switched-type network such as certain PBX networks 130. Rather, the primary purpose of the IP-CDMA gateway 155 is to compress the VoIP headers that accompany a packet of voice data and comprise control and overhead information so that the efficiencies of the limited bandwidth CDMA

network can be maximized as is discussed in greater detail below.

Like the voice terminals connected directly to the VoIP network 100, one or more of the voice terminals connected to the PBX and CDMA network gateways typically comprise a perceptual QoS agent for measuring and generating perceptual QoS metrics relative to the occurrence of perceptible anomalies in the voice stream as it is received. Alternatively, perceptual QoS test terminals (not shown) can be connected with each gateway or at strategic nodes within the PBX and CDMA networks 130 & 150. For instance, perceptual QoS agents, whether running on voice terminals or associated with test terminals, are typically provided for each cell within the CDMA network 150.

Referring back to Figure 1, a perceptual QoS database 180 is connected to the VoIP network. After each of the perceptual QoS agents connected with the VoIP network gathers QoS metric information, the metrics are transmitted to the QoS database where the metrics are stored along with associated information regarding the time the metrics were measured and the gateway or node with which the QoS agent was connected. In alternative embodiments, a number of databases can replace the single

database illustrated, wherein each gateway and/or each network node has its own database.

The perceptual QoS database 180 is utilized by one or more call admission controllers 185 that are either directly connected with the database as part of a call admissions center or can connect with the database over the VoIP network 100. The call admission controllers 185 calculate call admission thresholds that are utilized by each of the network nodes and/or gateways to determine whether to admit a new call onto the network through the node or gateway as is discussed in greater detail below. In alternative embodiments, call admission controllers 185 can be located directly at each node or gateway to control the call admissions at the associated gateway or node.

Figure 2 is a flow chart illustrating the operation of the VoIP network 100 concerning the admittance of a new call onto the network through a network gateway or node according to one embodiment of the present invention. In block 210, perceptual QoS agents located at various voice terminals distributed over the network measure and generate one or more perceptual QoS metrics that relate to perceptible variations in the quality of the voice stream. In the simplest form, the resulting metrics may comprise a single number indicating the overall quality

of the voice signal, or the metrics may comprise a more sophisticated set of data that individually describes the various anomalies within the voice signal, such as delay, choppiness, and speech level variation. The perceptual QoS agents can be in the form of software running in the background of a voice terminal or the agents can be in the form of hardwired circuits contained in the voice terminals. Alternatively, the agents can comprise dedicated perceptual QoS test terminals 120 that are connected to the VoIP network 100 through the various nodes and gateways of the network. Ideally, at least one perceptual QoS agent is connected to the network through each network node and/or gateway that also has voice terminals attached thereto.

Next, as shown in block 215, the perceptual QoS metrics gathered by each agent are transmitted to one or more perceptual QoS databases 180. In certain embodiments, a database can be resident in each node or gateway for storing metric information related only to those voice terminals connected to it. It is to be appreciated that the perceptual QoS metrics are measured, generated and transmitted to the perceptual QoS database(s) at predetermined intervals. The time span between the predetermined intervals may vary from a few

hundred milliseconds to a few hundred seconds depending on the particular network.

In block 220, the perceptual QoS metrics are analyzed by one or more call admission controllers 185 to determine the perceptual QoS level at each of the nodes or gateways. Next, in block 225, the call admission controller(s) 185 determine call admission threshold values for each portion of the network based on the perceptual QoS of the calls within a particular portion of the network served by a node or a gateway. The call admission threshold indicates a minimum perceptual QoS level below which new calls will not be serviced. For example, if the voice quality reported by perceptual QoS agents connected with IP-PBX gateway 140 in the VoIP network 100 indicates a low perceptual QoS level, a high threshold value may be generated, making it more difficult for new calls to gain access to the network through the IP-PBX gateway 140. It can be appreciated that as additional calls are admitted, the additional demands on the gateway 140 could cause the perceptual QoS levels of the active calls within this portion of the network to decrease to even lower levels. Ideally, the degradation of perceptual QoS levels below certain levels is to be avoided. In a preferred embodiment, the threshold value is set at a value between 0 and 1,

wherein higher values generally indicate an overall lower perceptual QoS level within an associated portion of the network. In another simpler embodiment, the threshold value at a node may be set to one number indicating an acceptable perceptual QoS and another number to indicate an unacceptable perceptual QoS.

In certain embodiments, the threshold values are solely based on perceptual QoS metrics that are measured in real time for each particular portion of the network. In other embodiments, the call admission controllers 185 may rely at least in part on historical data stored within the perceptual QoS databases 180 to predict future perceptual QoS levels within the portions of the network and set the call admission thresholds accordingly. In other embodiments, the call admission controllers 185 may utilize both real time perceptual QoS data and historical perceptual QoS data to generate call admission thresholds. Additionally, a call admission threshold for a particular portion of the network can be based in part on the perceptual QoS values in other parts of the network. It is to be appreciated that any number of models can be utilized to determine call admission thresholds for each of the various portions of the network as would be obvious to one of ordinary skill in the art with the benefit of this disclosure.

In the preferred embodiment, the threshold values are sent to each of the associated nodes and gateways for use by the nodes and gateways to determine whether to admit a new call on to the network.

In block 230, a call request is received by a node or gateway. Based on the call admission thresholds of the applicable portions of the network, the call is either admitted or blocked as indicated in blocks 245 and 240. Typically, the applicable call admission threshold(s) for a particular call request is compared to a priority value associated with the call request. The priority values typically comprise a number within a range similar to the range of threshold values. For instance, in the preferred embodiment, the priority values have a range from 0 to 1. If the priority value of the call request is equal to or greater than the applicable threshold values then the call is admitted. If the priority value is less than the call admission threshold, the requested call is blocked.

Any number of protocols can be established for assigning priority values to the call requests as can be appreciated by someone of skill in the art with the benefit of this disclosure. For instance in one embodiment, a call request is assigned a random number between 0 and 1, wherein the call is admitted if the

random number is higher than the associated threshold value(s). It is to be appreciated that when the traffic is light on the network and/or the QoS level being experienced by active users of the network is high, the threshold value will be low and the probability that the call will be admitted will be very high. For example, if the call threshold value for a node associated with a particular call request is 0.02, there will be a 98% chance that the requested call will be admitted.

In another embodiment, the call request may utilize a priority value associated with a particular voice terminal, wherein the voice terminals are assigned different priority values dependent on the relative importance of the users associated with the voice terminals. For example, in a VoIP network of a corporation, the executives of the corporation can be assigned very high priority values that greatly reduce the risk that their calls will be blocked. On the other hand, courtesy phones on a factory floor may have relatively low priority values such that it is less likely that calls from them will be admitted.

In yet another embodiment wherein the threshold value assigned to the node represents either an acceptable or unacceptable perceptual QoS on that node, no priority value may be assigned to the call request.

Rather, the call will be automatically admitted if the perceived QoS level is acceptable and block is the perceived QoS level on an associated node is not acceptable.

Typically, the VoIP networks of the present invention are designed to handle a certain volume of calls with an acceptable perceptual QoS. Accordingly, most of the time, the call admission thresholds for the various nodes and gateways on the network are extremely low and few, if any, calls are blocked. The call admission algorithms are therefore of particular importance when the capacity of the network or portions of the network is taxed. By using call admission algorithms based on perceptual QoS metrics that are directly related to the quality of the call as would be perceived by a user, the number of calls serviced by a portion of the network experiencing a high level of traffic can be maximized without causing unnecessary caller frustration due to poor call quality. This is in contrast to many prior art VoIP networks that assign hard limits to the various nodes and gateways indicating the maximum number of calls it will support. In certain situations, the hard limit may be lower than the number of calls that could have been supported at an acceptable quality level and in other instances, the hard limit may

be too high causing a situation wherein the quality of the active calls are below an acceptable level, potentially frustrating the users of the network.

It is to be appreciated that a typical call request on a VoIP network must have an associated priority value greater than call admission values associated with both the node or gateway with which the requesting voice terminal is connected, but also the node or gateway associated with the voice terminal that is to be receiving the requested call. If either call admission value is greater than the requested call's priority value, the call will be blocked.

Referring back to Figure 1, a VoIP over wireless (VoIPoW) network 150 can also be connected to the VoIP network as described above. A VoIPoW network 150 may be based on CDMA cellular technology, such as a third generation Wideband-CDMA (W-CDMA) technology. It is to be appreciated that as technology advances and other standards develop, a VoIPoW network may be based on those new standards as well. In addition to being used to determine call admission threshold values, perceptual QoS metrics are also used in CDMA networks (both VoIPoW and non-VoIPoW varieties alike) to dynamically maintain the perceptual QoS levels of active wireless calls by adjusting the transmission parameters of the base station

transceivers 165 and the mobile devices 170 & 175. The use of perceptual QoS metrics to actively and dynamically control the quality of CDMA network voice signals is described in detail in related patent applications: "System and Method for Quality Billing" (United States Patent Application No. 60/245,111 filed on November 1, 2000); and "System and Method for QoS Adaptive Communications" (United States Patent Application No. 60/240,530 filed on October 13, 2000) both owned by the assignee of this application, and incorporated herein by reference.

As described above, one or more mobile devices 170 & 175 including a perceptual QoS agent or a dedicated perceptual QoS test terminal generate perceptual QoS metrics for each base station transceiver 165 of the wireless network 150. Each perceptual QoS metric describes the perceived perceptual QoS for communications within an associated cell served by a particular base station transceiver 165. The metrics are transmitted to the base transceivers 165 and/or the CDMA mobile switching station 160, wherein the transmission characteristics of the CDMA network 150 are adjusted to provide and maintain satisfactory perceptual QoS levels for the active users of the wireless network. Additionally, the perceptual QoS metrics are transmitted

to perceptual QoS database 180. The call admission controller 185 then uses the perceptual QoS metrics to set the call admission thresholds for each of the base transceivers 165 of the CDMA VoIPoW network 150. The call admission thresholds are then sent back to the mobile switching center 160 and the base transceivers 165, as applicable, for use in determining whether to admit call requests associated with wireless voice terminals.

Two of the greatest challenges in porting VoIP networks to a wireless environment has been to both maintain an acceptable perceptual QoS level and maintain the capacity to service a suitable number of users simultaneously. The CDMA VoIPoW network 150 is constrained by the limited capacity of each base station transceiver 165, especially when the configuration of each VoIP packet is considered. Each VoIP packet typically comprises a 40 byte header containing central and overhead information regarding the routing of the packet and the reassembly of voice data within the packet with the voice data of other packets. In contrast, the size of the voice data contained within each packet can be as small as 10 bytes and is typically smaller than the size of an uncompressed header. Accordingly, during a typical transfer of voice data over a VoIPoW CDMA network utilizing uncompressed headers substantially more than

50% of the networks capacity is used to transport overhead and control information. Since the overhead and control information greatly reduces the capacity of the CDMA network, header compression algorithms have been developed that significantly reduce the size of the header.

Compression algorithms take advantage of a high degree of redundancy in the fields of the headers of consecutive packets. In general, after a packet with an uncompressed header has been transmitted, subsequent headers need only contain information about how its context has varied relative to the header in the preceding packet. Certain well known compression algorithms can reduce the size of a 40 byte packet to as low as 1 byte.

Highly compressed headers present problems when utilized with error prone wireless connections. When packets are lost or damaged during transport over the wireless network, downstream packets may not be able to properly recreate their headers at the voice terminal thereby introducing errors into the voice stream. Most compression algorithms have mechanisms to repair problems related to the loss of context between frames, but the time involved in repairing the frames may be too great in a real time transport environment and often a significant

number of frames must be discarded. It should be appreciated that the loss of any significant number of consecutive frames has a measurable effect on a user's perceptual QoS.

The dilemma in a VoIPoW environment is therefore, whether to (i) ensure a high perceptual QoS by transmitting IP packets with headers that have not been compressed, reducing the VoIPoW networks capacity significantly, or (ii) increase the overall capacity of the network, potentially reducing the perceptual QoS of the users of the VoIPoW network due to packets that are lost or damaged in transport.

Different header compression algorithms can be utilized that balance the amount of header compression with the possible loss of packet data in a real time environment. For example, if maximum header compression is desired to maximize the user capacity of a VoIPoW network, the first packet transmitted to the mobile device 170 or 175 from the base station transceiver 165 would contain an uncompressed header and all subsequent headers would be compressed with each subsequent compressed header being based on the change in context relative to the previous header. If a packet is lost during transmission, all subsequent packets would be lost until the mobile device requests and receives a new

uncompressed header from the base station transceiver to reestablish the context between headers.

The percentage of voice data transmitted in each packet when a highly compressed header is utilized is 83% or greater relative to the total size of the packet (when a 2 byte compressed header is utilized with a 10 byte voice data segment, the voice data to total transmitted data ratio would be 83.3%). On the other hand if no compression is utilized in order to maximize the QoS of the VoIPoW network, the percentage of voice data transmitted in each packet relative to the total size of the packet would be as low as 20% (assuming a 40 byte header and a 10 byte voice data segment). Intermediate compression algorithms wherein a packet with an uncompressed header is transmitted in between packets with compressed headers at regular intervals may also be utilized. It can be appreciated that the periodic transmission of uncompressed headers will reduce the overall average capacity of a base transceiver, but will minimize the number of consecutive packets that must be discarded if a packet with a compressed header is lost or damaged.

In a preferred embodiment of the present invention, the compression algorithm utilized in relation to a base station transceiver 165 of a VoIPoW CDMA network 150 is

varied or changed based on the perceptual QoS levels of mobile devices or perceptual QoS test terminals actively connected with the base transceiver. For instance, if the perceptual QoS levels drop below a suitable level, the level of header compression may be reduced to ensure that fewer packets of voice data have to be discarded because of packets lost or damaged during transmission. In another instance, if QoS level relative to a base transceiver is high, the header compression level may be increased so that more mobile devices may be simultaneously supported by the base station transceiver 165.

In certain embodiments, the header compression levels for a base station transceiver 165 may be considered in conjunction with the perceptual QoS levels of the base transceiver to determine the call admittance threshold for the base transceiver. In this regard, a base transceiver having a high perceptual QoS and utilizing a high level of header compression would have a different call admission threshold than another base transceiver 165 having a similar perceptual QoS level but utilizing a much lower amount of header compression.

The present invention has been described in terms of voice over internet protocol networks; however, it is understood that similar perceptual QoS algorithms and

mechanisms can be utilized in the transport of other multimedia signals such as real time video streaming for use in video conferencing, or real time audio streaming. Additionally, other protocols may be implemented into a VoIP network that allow the admission and transport of packets over the network that are not related to real time voice communication. For instance, the call admission controller could also issue admittance threshold values to the various nodes, gateways and base station transceivers based on the type of data being transmitted. For example, packets containing email data can have a very low admittance threshold but be given a very low routing priority based on available capacity.

Furthermore, the present invention has been described primarily in terms of an integrated VoIP network wherein a VoIPoW network is integrated thereto. It is to be appreciated that the VoIPoW network could be a standalone network that is not actively integrated with a traditional wired VoIP network. In such an embodiment, the call admission controller and the perceptual QoS database would be connected with or an integral part of the mobile switching center 160. It is further understood that in alternative embodiments, the call admission controllers and databases maybe integrated directly into the various gateways and network nodes,

wherein the perceptual QoS metrics for a node are gathered and analyzed by the node to generate threshold values that it utilizes to determine whether to admit a call within its portion of the network. In yet other embodiments, the call admission controller may make call admission decisions instead of the respective nodes and gateways. In this regard, expert systems may be utilized to manage the perceptual QoS levels on the network as a whole to ensure both that an acceptable perceptual QoS level is maintained among all active users while maximizing the probability that new call requests will be admitted.

Although the present invention has been described with a certain degree of particularity, it is understood that the present disclosure has been made by way of example, and changes in detail or structure may be made without departing from the spirit of the invention as defined in the appended claims.

The present application claims priority on a provisional application, U.S. Serial No. 60/240,535, entitled "System and Method for QoS-Based Call Admission for VoIP, VoIPoW and CDMA Systems", filed on October 13, 2000, which is incorporated by reference herein.

(Industrial Applicability)

This invention is applied generally to Voice over IP (VoIP) and Mobile communications networks, and more specifically to methods and apparatus for determining whether to admit a new call onto a network based on the perceptual quality of service of other communications active on the network, as well as the required perceptual quality level by the user requesting the new call.

CLAIMS

1. A method of operating a communications network comprising the steps of:

establishing a connection between a first receiving device and the communications network through a first node;

transmitting a packetized signal over the communications network through the first node to the first receiving device; and

determining the perceptual quality of the packetized signal by measuring occurrences of one or more perceptible anomalies within the packetized signal as it is received at the first receiving device in real time.

2. The method according to claim 1, wherein the first receiving device is selected from the group consisting of: a dedicated perceptual quality of service (QoS) measurement apparatus, a telephone and a personal computer.

3. The method according to claim 1, wherein the first receiving device receives a signal selected from the group consisting of: an electronic data signal, a voice signal, an audio signal and a video signal.

4. The method according to claim 1, wherein the communications network is a VoIP network.

5. The method according to claim 1, wherein the node is a gateway terminal between the communications network and another communications network.

6. The method according to claim 1, wherein the communication network is selected from the group consisting of: a PBX network, a VoIP network and a VoIPoW network.

7. The method according to claim 1, wherein the said perceptible anomalies are selected from the group consisting of: audio distortion, audio choppiness, audio jitter and variations in audio level.

8. The method according to claim 1, further comprising the step of:

generating an admission threshold value for the first node based on the perceptual quality of service level, said admission threshold value for use by the first node to determine whether to admit a new signal onto said communications network through said first node.

9. The method according to claim 8, further comprising the steps of:

receiving a request for admittance of a new signal at said first node;

comparing the admission threshold value with a priority number associated with said request; and

admitting said new signal based on said comparison.

10. A method for controlling call admission on a VoIP network, said method comprising the steps of:

transmitting a voice stream from an active call to a receiving device over the VoIP network through a network node;

generating perceptual quality of service (QoS) metrics for said network node by measuring the presence of perceptible variations in said voice stream at predetermined intervals at a communications device connected with the VoIP network through said network node;

generating a call admission threshold value for said network node based on the perceptual QoS metrics;

receiving a request to admit a new call onto the VoIP network through said network node;

comparing said call admission threshold value with a priority number associated with the request; and

admitting the new call based on said comparison.

11. The method according to claim 10, wherein the priority number of said request is derived from one or more priority numbers that are associated with one or more communications devices associated with the request to admit a new call.

12. The method according to claim 11, wherein each of the one or more communications devices are assigned a priority number.

13. The method according to claim 10, wherein said priority number of the request is randomly assigned when the call request is initiated.

14. The method according to claim 10, wherein said call admission threshold value and the priority number each have a numerical value between 0 and 1.

15. The method according to claim 14, wherein said admitting the new call based on the comparison further comprises the step of:

admitting the new call if said priority number is greater than said call admission threshold value.

16. The method according to claim 10, further comprising the step of:

transmitting the perceptual QoS metrics to a perceptual QoS database, said perceptual QoS database containing perceptual QoS metrics data from said network node.

17. The method according to claim 10 wherein said perceptual QoS database distributes corresponding admission threshold values to a plurality of nodes within said VoIP network.

18. A VoIP network comprising:

a plurality of routers connected together for the transmission of voice signal packets therebetween;

a plurality of nodes, said nodes being connected to at least one of said routers; and

a plurality of communications devices connected to said plurality of nodes, each of said communication devices including a perceptual quality of service (QoS) agent, the perceptual quality of service agent configured to measure at predetermined intervals perceptible variations in a voice stream received by the respective

communications device and generate perceptual QoS metrics based on the perceptible variations.

19. The VoIP network according to claim 18, further comprising a perceptual QoS database, said perceptual QoS database connected to at least one of said routers and configured to store said perceptual QoS metrics generated by each perceptual QoS agent of each communications device of said plurality of communications devices.

20. The VoIP network according to claim 18, further comprising a call admission controller, said call admission controller connected to at least one of said routers, configured to generate new call admission threshold values for each node of said plurality of nodes based on said perceptual QoS metrics.

21. The VoIP network according to claim 20, wherein said plurality of nodes determines whether to admit a new call based in part on the call admission threshold values associated with said network node.

22. The VoIP network according to claim 20, wherein said call admission controller transmits said call

admission threshold values to at least one of said plurality of nodes.

23. The VoIP network according to claim 18, further comprising a gateway, said gateway being connected to a switched circuit type network.

24. The VoIP network according to claim 23, wherein said switched circuit-type network is a private branch exchange (PBX).

25. The VoIP network according to claim 20, further comprising a perceptual QoS database being connected with said call admission controller, said perceptual QoS database configured to store said perceptual QoS metrics generated by said perceptual QoS agents.

26. The VoIP network according to claim 18, wherein said plurality of communications devices are selected from the group consisting of: a telephone, a personal computer and a dedicated perceptual QoS measurement device.

27. The VoIP network according to claim 18, wherein said perceptual QoS agent comprises software executable on said communications device.

28. A method of operating a wireless network, said method comprising the steps of:

measuring occurrences of one or more anomalies within a packetized data signal transmitted over said wireless network;

receiving a first internet protocol (IP) packet, said IP packet comprising a payload section and a header, said header comprising overhead information; and

compressing said overhead information based at least partly on the occurrences of one or more anomalies.

29. The method according to claim 28, wherein said one or more anomalies are perceptible variations of said packetized data signal.

30. The method according to claim 29, wherein said packetized data signal comprises a voice signal and said payload section comprises voice data.

31. The method according to claim 28, wherein said one or more anomalies are measured in real time.

32. The method according to claim 28 further comprising the step of: decreasing the compression of the overhead information if the occurrences of said one or more anomalies increases from a first level to a second level.

33. The method according to claim 28 further comprising the step of: increasing the compression of the overhead information if the occurrences of one or more anomalies decreases from a first level to a second level.

34. The method according to claim 30 further comprising the step of: measuring occurrences of said one or more anomalies by a wireless device receiving said voice signal, said wireless device being in communications with a base station transceiver over a wireless connection.

35. The method according to claim 28 wherein changes to the compression of the packets are based upon algorithms that are adaptively modified by an intelligent network based upon network conditions.

36. The method according to claim 35 wherein network conditions are QoS levels.

37. The method according to claim 28, wherein the wireless network is a code division multiple access (CDMA) network.

38. A method for controlling call admission on a VoIPoW network using code division multiple access technology, said method comprising the steps of:

determining the presence of one or more perceptible variations in a voice stream received by a first wireless device in wireless communication with a base station transceiver;

generating a perceptual QoS metric based on said one or more perceptible variations;

determining a call admission threshold for said base station transceiver based on the perceptual QoS metric; and

determining whether to admit a new call through said base station receiver based on said call admission threshold value.

39. The method according to claim 38, further comprising the steps of:

receiving a request to initiate a new wireless call by establishing a wireless connection between said base station transceiver and a second wireless device; and

determining whether to admit said new wireless call based on said call admission threshold for the base station transceiver.

40. The method according to claim 38, further comprising the step of:

receiving a first internet protocol (IP) packet, said IP packet comprising a payload section and a header, said header comprising overhead information; and

compressing said overhead information based at least partly on the occurrences of one or more anomalies.

41. The method according to claim 40 wherein changes to said compression of said packets are based upon algorithms that are adaptively modified by an intelligent network based upon network conditions.

42. The method according to claim 41 wherein changes to said network conditions are perceptual QoS Levels.

43. The method according to claim 38, further comprising the step of:

adjusting resources of said VoIPoW network as necessary based on said perceptual QoS metric to maintain an acceptable perceptual QoS level for said first wireless device.

44. The method according to claim 38, further comprising the steps of:

assigning a priority value to said second wireless device;

comparing the threshold value with the priority value; and

admitting the call if the priority value is greater than the threshold value.

45. In a communications network, a system for perceptual QoS based call admission, said system comprising:

a plurality of routers connected together for the transmission of signals therebetween;

a plurality of nodes, said nodes being connected to at least one of said routers;

a plurality of communications devices connected to said plurality of nodes, each of said communication

devices including a perceptual quality of service (QoS) agent, said perceptual quality of service agent configured to measure at predetermined intervals perceptible variations in said signal received by the respective communications devices and generate perceptual QoS metrics based on the perceptible variations;

a threshold means for generating a call admission threshold value for said network nodes based on said perceptual QoS metrics;

a receiving means for receiving a request to admit a new call onto said network through said plurality of the network nodes;

a comparing means for comparing said call admission threshold value with a priority number associated with said request; and

an admission means for admitting said the new call based on said comparison.

46. The call admission system according to claim 45, wherein said network is selected from the group consisting of: a PBX network, a VoIP network and a VoIPoW network.

47. The call admission system according to claim 45, wherein said plurality of communications devices is

selected from the group consisting of: a telephone, a personal computer and a dedicated perceptual QoS measurement device.

48. The call admission system according to claim 45, wherein said variations in a signal is selected from the group consisting of: audio distortion, audio choppiness, audio jitter and variations in audio level.

49. The call admission system according to claim 45, wherein said signals are comprised of internet protocol (IP) packets, said IP packets comprising a payload section and a header, said header comprising overhead information.

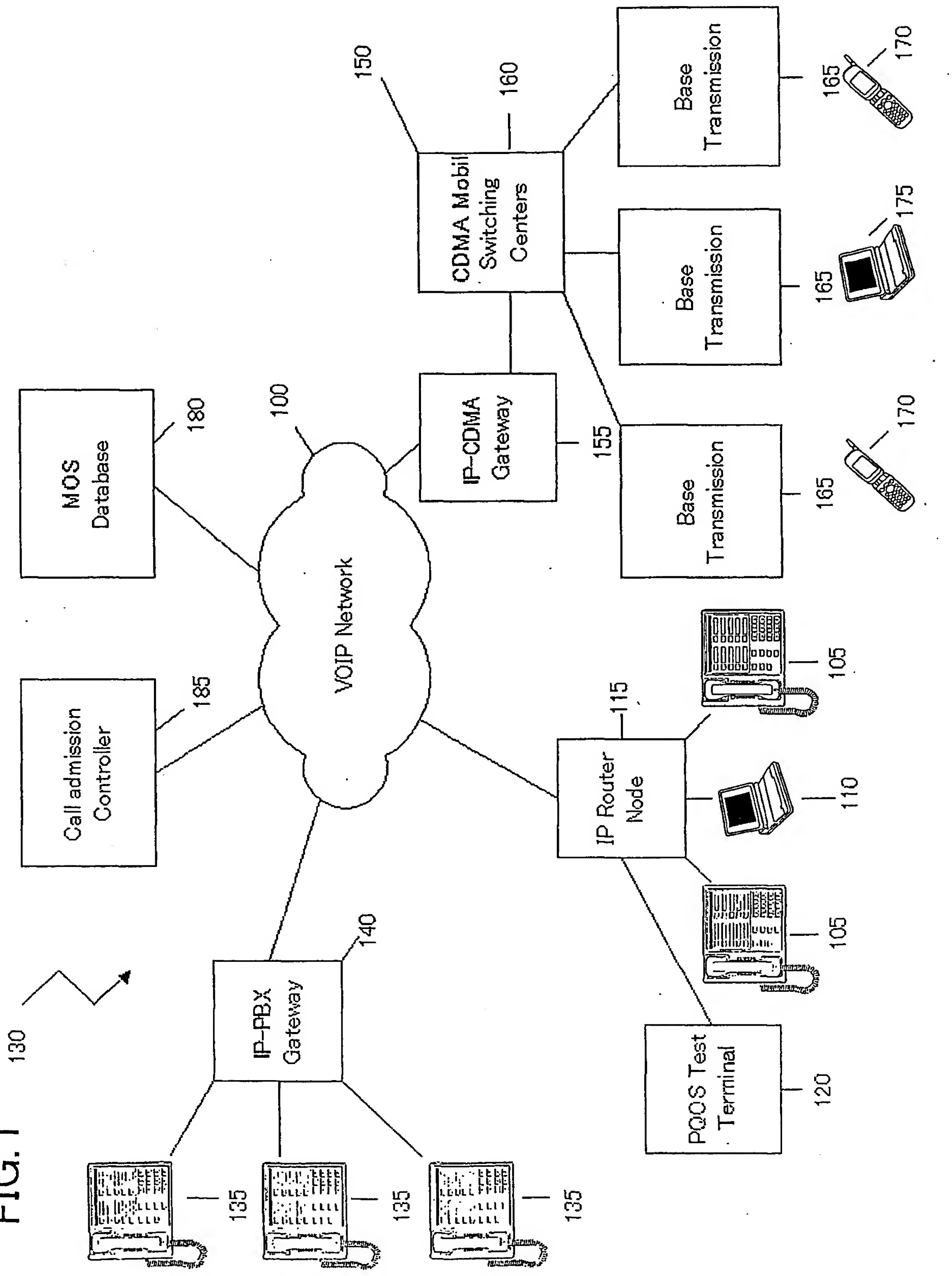
50. The call admission system according to claim 49 further comprising:

a compression means for compressing said overhead information based at least partly on said variations in said signal.

51. The call admission system according to claim 50, whereby said compression means is decreased if the occurrences of said variations in said signal increases from a first level to a second level.

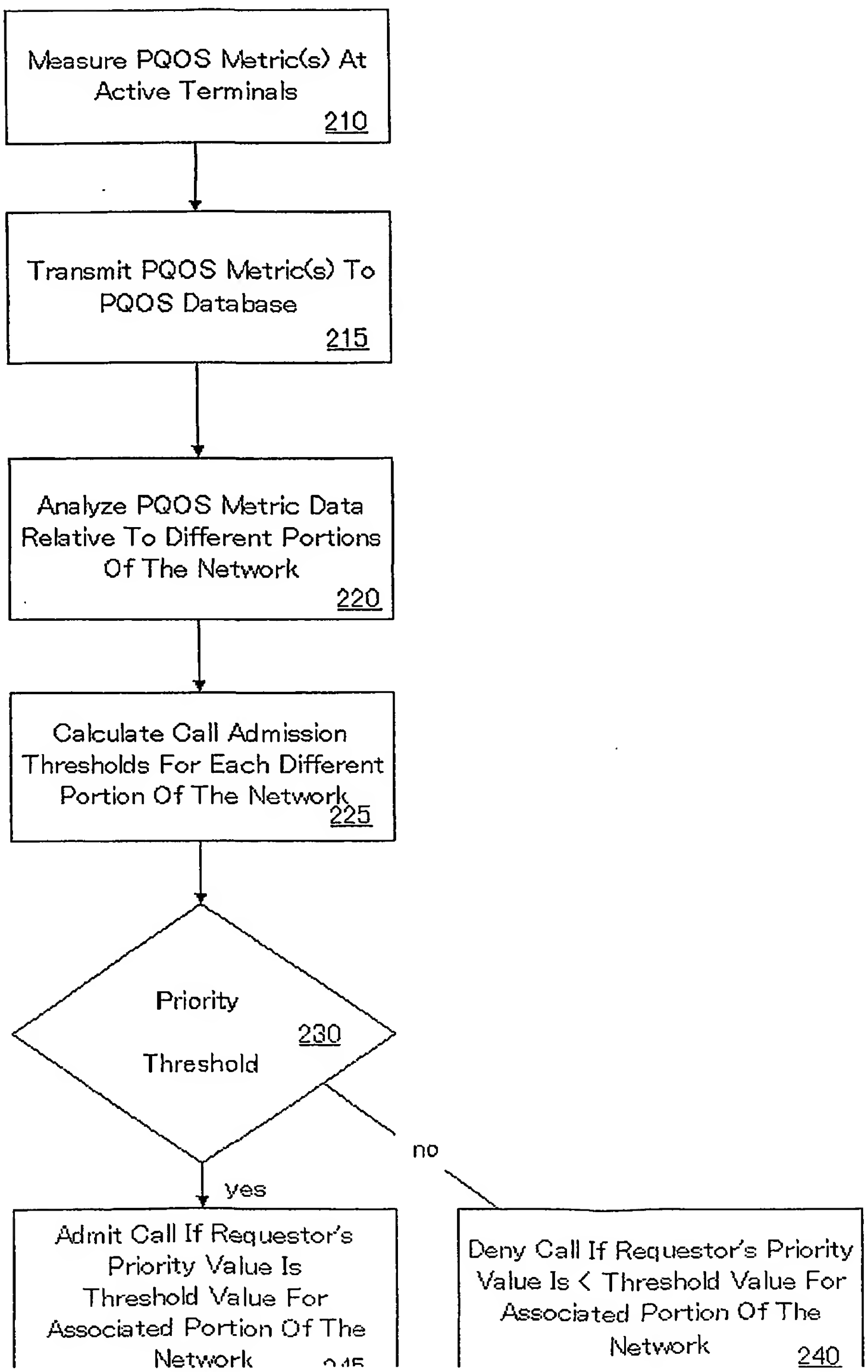
52. The call admission system according to claim 50, whereby said compression means is increased if the occurrences of said variations in said signal decrease from a first level to a second level.

FIG.1



130
MOS Database
180
100
VOIP Network
Call admission Controller
185
IP-PBX Gateway
140
135
135
135
IP-CDMA Gateway
155
CDMA Mobil Switching Centers
150
160
Base Transmission
165
170
175
IP Router Node
115
105
110
105
PQOS Test Terminal
120

FIG.2


Measure PQOS Metric(s) At Active Terminals 210
Transmit PQOS Metric(s) To PQOS Database 215
Analyze PQOS Metric Data Relative To Different Portions Of The Network 220
Calculate Call Admission Thresholds For Each Different Portion Of The Network 225
Priority 230 Threshold
no
yes
Admit Call If Requestor's Priority Value Is Threshold Value For Associated Portion Of The Network
Deny Call If Requestor's Priority Value Is < Threshold Value For Associated Portion Of The Network 240